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Harmonizacija telekomunikacij in internetnega protokola prek omrežij (TIPHON), 3. izdaja - Kakovost storitve od konca do konca v sistemih TIPHON - 6. del: Dejanske meritve parametrov karakteristik in zmogljivosti omrežja in terminalov v omrežjih TIPHON ter njihov vpliv na kakovost govora

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 6: Actual measurements of network and terminal characteristics and performance parameters in TIPHON networks and their influence on voice quality

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Technical Report

Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 6: Actual measurements of network and terminal characteristics and performance parameters in TIPHON networks and their influence on voice quality

Reference

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Foreword

This Technical Report (TR) has been produced by ETSI Project Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON).

The present document is part 6 of a multi-part deliverable covering End-to-end Quality of Service in TIPHON systems, as identified below:

TR 101 329-1: "General aspects of Quality of Service (QoS)";

TS 101 329-2: "Definition of speech Quality of Service (QoS) Classes";

TS 101 329-3: "Signalling and control of end-to-end Quality of Service (QoS)";

TS 101 329-5: "Quality of Service (QoS) measurement methodologies";

TR 101 329-6: "Actual measurements of network and terminal characteristics and performance parameters in TIPHON networks and their influence on voice quality";

TR 101 329-7: "Design guide for elements of a TIPHON connection from an end-to-end speech transmission performance point of view".

Quality of Service aspects of TIPHON Release 4 and 5 Systems will be covered in TS 102 024 and TS 102 025 respectively, and more comprehensive versions of the Release 3 documents listed above will be published as part of Release 4 and 5 as work progresses.

Introduction

The present document forms one of a series of technical specifications and technical reports produced by TIPHON Working Group 5 addressing Quality of Service (QoS) in TIPHON Systems. The structure of this work is illustrated in Figure 1.

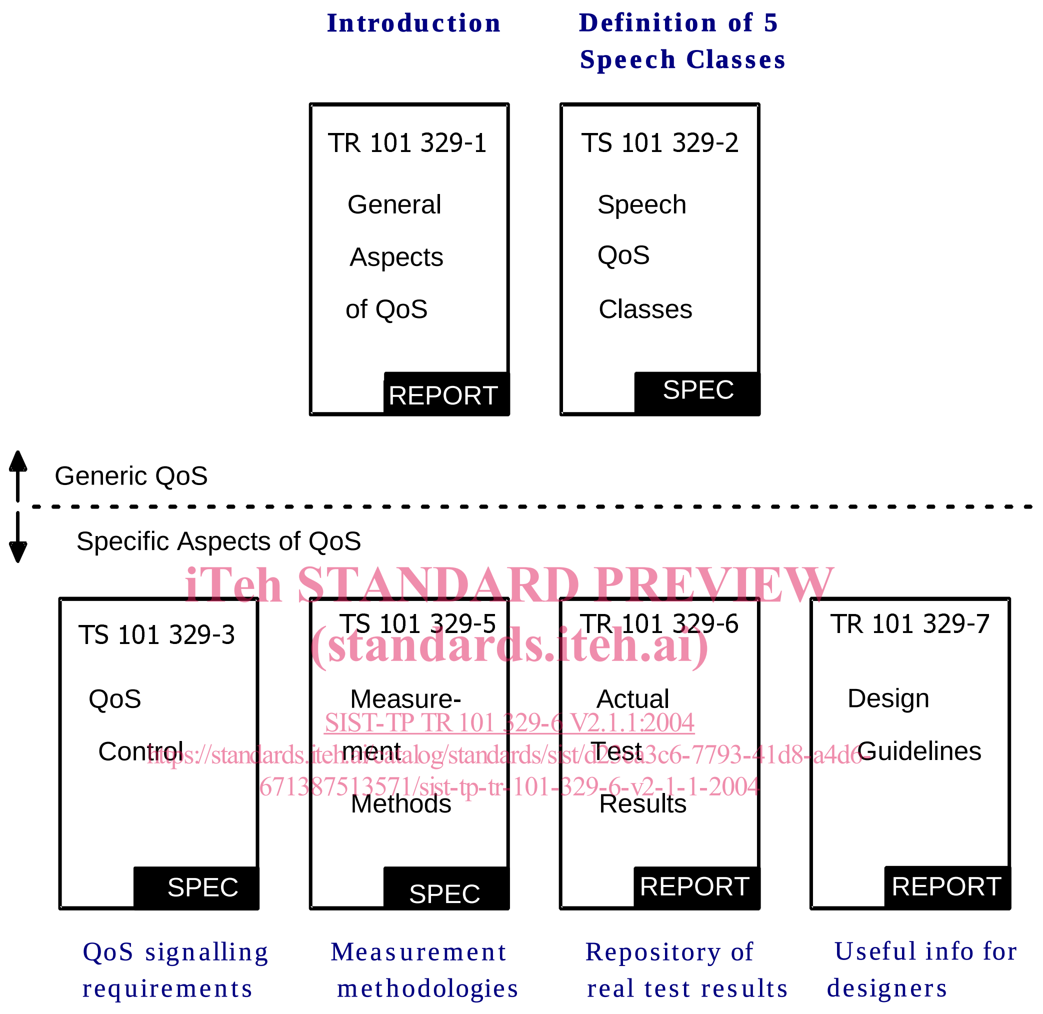


Figure 1: Structure of TIPHON QoS Documentation for Release 3

1 Scope

The present document applies to IP networks that provide voice telephony in accordance with any of the TIPHON Scenarios.

The objective with the present document is to collect all results of various VoIP speech transmission quality tests and related information. This collection should be used for information and to review and discuss the values of the TIPHON QoS classes which are described in WG5 documents TR 101 329-1 [3] and TR 101 329-7 [6].

The separate measurements should give a very good opportunity to understand the goal of the measurement itself and the exact measurement Set Up conditions to understand under which framework the measurements were done.

The present document covers measurement results provided to TIPHON during the years 1999 to 2001, which have contributed to the measurement methodologies in 101 329-5 [5] as well as providing design parameters in 101 329-7 [6].

2 References

For the purposes of this Technical Report (TR) the following references apply:

[1] ETSI ETR 275 (1996): "Transmission and Multiplexing (TM); Considerations on transmission delay and transmission delay values for components on connections supporting speech communication over evolving digital networks".

[2] ETSI TR 101 329 (V2.1.1): "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON); General aspects of Quality of Service (QoS)".

[3] ETSI TR 101 329-1: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 1: General aspects of Quality of Service (QoS)".

[4] ETSI TR 101 329-2: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 2: Definition of speech Quality of Service (QoS) classes".

[5] ETSI TR 101 329-5: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 5: Quality of Service (QoS) measurement methodologies".

[6] ETSI TR 101 329-7: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 7: Design guide for elements of a TIPHON connection from an end-to-end speech transmission performance point of view".

[7] ETSI ES 201 168 (V1.1.1): "Corporate Telecommunication Networks (CN); Transmission characteristics of digital Private Branch eXchanges (PBXs)".

[8] ETSI GTS 06.10 (V3.2.0): "European digital cellular telecommunications system (Phase 1); GSM Full Rate Speech Transcoding (GSM 06.10)".

[9] ITU–T Recommendation G.107: "The E-Model, a computational model for use in transmission planning".

[10] ITU–T Recommendation G.108: "Application of the E-model: A planning guide".

[11] ITU–T Recommendation G.109 (1999): "Definition of categories of speech transmission quality".

[12] ITU–T Recommendation G.113 (2001): "Transmission impairments due to speech processing".

[13] ITU–T Recommendation G.131 (1996): "Control of talker echo".

[14] ITU–T Recommendation G.165 (1993): "Echo cancellers".

[15] ITU–T Recommendation G.168 (2000): "Digital network echo cancellers".

[16] ITU–T Recommendation G.711 (1988): "Pulse code modulation (PCM) of voice frequencies".

[17] ITU–T Recommendation G.721 (1988): "32 kbit/s adaptive differential pulse code modulation (ADPCM)".

[18] ITU–T Recommendation G.723.1 (1996): "Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s".

[19] ITU–T Recommendation G.726 (1990): "40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)".

[20] ITU–T Recommendation G.727 (1990): "5-, 4-, 3- and 2-bit/ sample embedded adaptive differential pulse code modulation (ADPCM)".

[21] ITU–T Recommendation G.728 (1992): "Coding of speech at 16 kbit/s using low-delay code excited linear prediction".

[22] ITU–T Recommendation G.729 (1996): "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)".

[23] ITU–T Recommendation G.729A (Annex A - 1996): "Reduced complexity 8 kbit/s CS-ACELP speech codec".

[24] ITU–T Recommendation G.729B (Annex B - 1996): "A silence compression scheme for G.729 optimized for terminals conforming to Recommendation V.70".

[25] ITU–T Recommendation H.323 (2000): "Packet-based multimedia communications systems".

[26] ITU-T Recommendation P.57: "Artificial ears".

[27] ITU-T Recommendation P.58: "Head and torso simulator for telephonometry".

[28] ITU–T Recommendation P.64 (1999): "Determination of sensitivity/frequency characteristics of local telephone systems".

[29] ITU-T Recommendation P.501: "Test signals for use in telephonometry".

[30] ITU-T Recommendation P.502: "Objective test methods for speech communication systems, using complex test signals".

[31] ITU-T Recommendation P.581: "Use of head and torso simulator (HATS) for hands-free terminal testing".

[32] ITU–T Recommendation P.800 (1996): "Methods for subjective determination of transmission quality".

[33] ITU–T Recommendation P.861 (1998): "Objective quality measurement of telephone-band (300-3 400 Hz) speech codecs".

[34] ITU–T Recommendation Q.13/12: "Rapporteur of Question 13 inside ITU-T Study Group 12; ETSI TIPHON 17TD135".

[35] SG 16, Santiago, Chile, 17-28 May 1999; D.249 (WP 3/16): "A High Quality Low-Complexity Algorithm For Frame Erasure Concealment (FEC) With G.711" (Source: AT&T).

[36] T1A1.7/99-012r3; Jul-28-1999: "Draft Proposed American National Standard - A Packet Loss Concealment Technique for Use with ITU-T Recommendation G.711" (Source: AT&T).

[37] ETSI ETS 300 581-2: "Digital cellular telecommunications system (Phase 2) (GSM); Half rate speech; Part 2: Half rate speech transcoding (GSM 06.20 version 4.3.1)".

[38] ETSI EN 300 726: "Digital cellular telecommunications system (Phase 2+) (GSM); Enhanced Full Rate (EFR) speech transcoding (GSM 06.60 version 8.0.1 Release 1999)".

[39] ITU-T Recommendation P.862 (2001): "Perceptual evaluation of speech quality (PESQ), an objective method for end-to-end speech quality assessment of narrowband telephone networks and speech codecs".

[40] ITU-T Recommendation G.723: "Extensions of Recommendation G.721 adaptive differential pulse code modulation to 24 and 40 kbit/s for digital circuit multiplication equipment application".

[41] ITU-T Recommendation G.723.1-A: "Speech coders : Silence compression scheme".

[42] ITU-T Recommendation P.50: "Artificial voices".

[43] ITU-T Recommendation G.114: "One-way transmission time".

3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACR	Absolute Category Rating
ASL	Active Speech Level
CAS	Communication Analysis System (HEAD acoustics test system)
CSS	Composite Source Signal
EC	Echo Canceller
EP	Error Pattern
GSM	Global System for Mobile communications
GSM EFR	GSM Enhanced Full Rate Speech Coder
GSM FR	GSM Full Rate Speech Coder
HATS	Head And Torso Simulator
IP	Internet Protocol
IRS	Intermediate Reference System
ISDN	Integrated Services Digital Network
JLR	Junction Loudness Rating
LAN	Local Area Network
MNRU	Modulated Noise Reference Unit
MOS	Mean Opinion Score
NLP	Non-Linear Processor
OLR	Overall Loudness Rating
OVL	Over-Load Point
PESQ	Perceptual Evaluation of Speech Quality (see ITU-T Recommendation P.862)
PLC	Packet Loss Concealment
PSTN	Public Switched Telephone Network
PVS	PC Voice Switch
QoS	Quality of Service
RLR	Receive Loudness Rating
SCN	Switched Communications Network
SLR	Send Loudness Rating
TMOS	TOSQA Mean Opinion Score (output of TOSQA)
TOSQA	Telecommunication Objective Speech Quality Assessment
VAD	Voice Activity Detection

4 List of Measurement Results

Table 1: List of measurement results

Nr.	Document	Source	Document Introduction	Date
1	Simulation Results of VoIP scenarios	Deutsche Telekom Berkom t.scheerbarth@berkom.de; i.kliche@berkom.de	ETSI TIPHON 11TD064	11/01/1999
2	APPENDIX I (to ITU-T Recommendation G.113 [12]	Mark E. Perkins mperkins@att.com	ETSI TIPHON 11TD084	11/01/1999
3	Speech Quality Test results of IP equipment in a LAN environment	Robert Bosch GmbH Joachim.Pomy@Tenovis.com	ETSI TIPHON 14TD081	16/07/1999
4	QoS Measurements of IP-Configurations	HEAD acoustics, Robert Bosch GmbH, T-Nova (Deutsche Telekom) h.w.gierlich@head-acoustics.de	ETSI TIPHON 15TD089	05/10/1999
5	Subjective Results on impairment effects of IP packet loss	Nortel Networks paulcov@nortelnetworks.com	ETSI TIPHON 17TD167	14/03/2000
6	Subjective and Objective Speech Quality Evaluation on Speech Data recorded at the SuperOp 99 event in Hawaii	Rapporteur of ITU-T Recommendation Q.13/12 [34]	ETSI TIPHON 17TD135	15/03/2000
7	Anonymous Test report of ETSI Speech Quality Test Event 2000	Deutsche Telekom, T-Nova; HEAD acoustics	ESTI TIPHON 22TD38	26/03/2001
8	Problems with the behaviour of jitter buffers and their influence on the end-to-end speech quality	Pieter Veenstra p.k.veenstra@kpn.com	ESTI TIPHON 22TD47	26/03/2001

5 General Measurement Results

5.1 Subjective Testing

5.1.1 Simulation Results of VoIP Scenarios

Source: Deutsche Telekom Berkom; Simulation Results of VoIP Scenarios; ETSI TIPHON 11TD064.

5.1.1.1 Introduction

ETSI TIPHON WG 5 has defined a methodology for testing VoIP End-to-End speech quality. This methodology was used as a basis model for the T Berkom simulation processing. Figure 2 shows the methodology used for simulation.

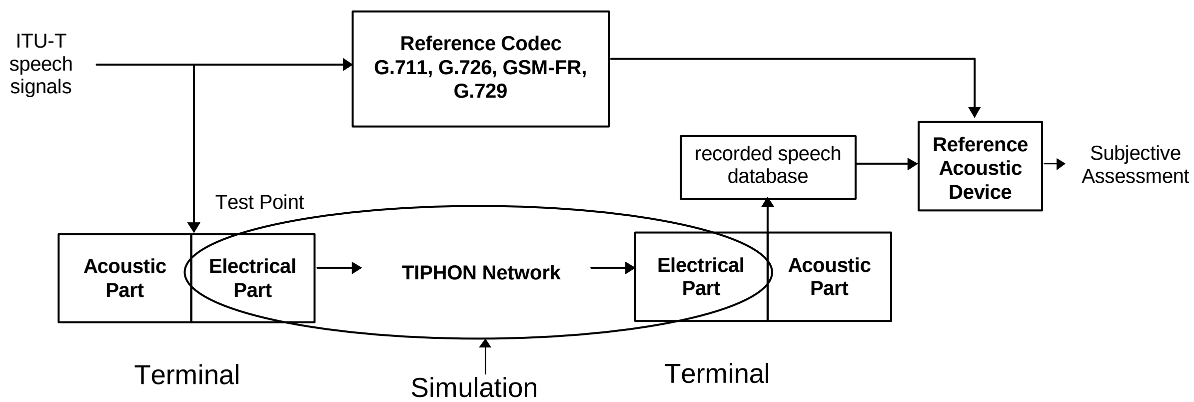


Figure 2: Simulation methodology for testing TIPHON speech quality

A set of speech signals designed according to ITU-T Recommendation P.800 [32] was used as input of the simulation path. The simulation path includes the terminal side (electrical part) and the network itself. The influence of the terminal side was focussed to the speech conversion and IP packet size issue. The influence of the network side was simulated by different packet loss rates.

After the simulation the speech samples were recorded and stored in a database.

The subjective assessment was carried out according to the ITU-T Recommendation P.800 [32] method.

5.1.1.2 Measurement Set Up

5.1.1.2.1 Basics

All source speech samples consisted of German sentences spoken by four talkers (2 male, 2 female). The input level taken for all scenarios was ASL = -26 dB Ovl. In the pre- and post-processing phase the speech samples were filtered with the modified IRS transmit and receive filter.

For every test condition the speech file was encoded and then assembled in IP packets. These IP packets were assembled with different lengths, according to the concerned speech frame number per packet.

For simulation of network influences in the case of packet loss, a common channel model was designed, realized by channel files which describe the network condition with the same time resolution as the source speech sample rate. So the network has a certain condition (good or bad) for every speech sample (every 125 µs), two adjacent network states were considered as statistically independent because the network speed was assumed to be much higher than the sample rate (8 000 samples per second). So for each packet loss rate one channel file was created using a random generator. The length of this channel file was exactly the same as the length of the speech file.

In a further step the speech file, assembled in IP packets, was matched to the channel file. According to the length of the IP packet (10 ms, 20 ms,...) the channel file was checked every time when a packet was ready to send. That means if the packet size was 10 ms the channel file was checked also every 10 ms if the condition is good or bad. In a bad case the IP packet was lost, otherwise it was further processed.

This information (IP packet lost or not) was stored in a description file which was the input of the re-assembler and speech decoder.

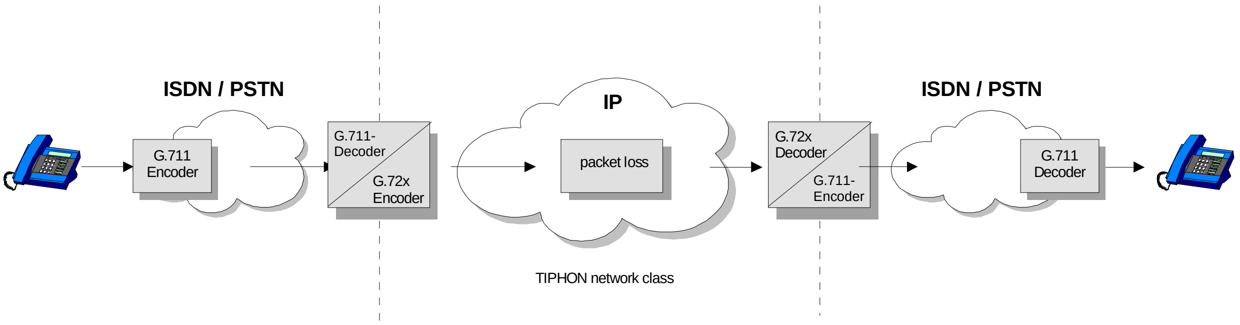
5.1.1.2.2 Test Cases

The test cases consisted a group of single codec scenarios (references), phone to phone scenarios in fixed network environments and a group of tandeming conditions. The tandeming conditions based on real scenarios where a mobile customer is connected to an ordinary telephone via IP. For this cases the GSM Full Rate Codec (GSM-FR) and the GSM Enhanced Full Rate Codec (EFR) were used. In such scenarios mainly the influence of the IP network was taken into account. Only one condition was chosen to simulate a voice transmission from a mobile phone to an ordinary telephone via an impaired radio channel and via an IP network with packet loss.

Table 2: G.711 [16] + Codec + G.7xx (Phone to Phone Scenario in fixed network environments)

Codec	Packet Loss	Speech Frame Size	Nr. of Frames/Packet	Substitution
G.711 [16]	5 %, 10 %, 15 %, 20 %	0,125 ms	80, 320, 480, 800	Silence
G.729B [24]	5 %	10 ms	1, 4, 6, 10	G.729 [22] internal
G.723.1 [18] (5.3)	0 %, 5 %, 10 %, 15 %, 20 %	30 ms	1, 2, 3	G.723.1 [18] internal
G.728 [21]	0 %, 5 %, 10 %, 15 %, 20 %	0,625 ms	16, 64, 96, 160	proprietary

logical scenario:



simulation scenario:

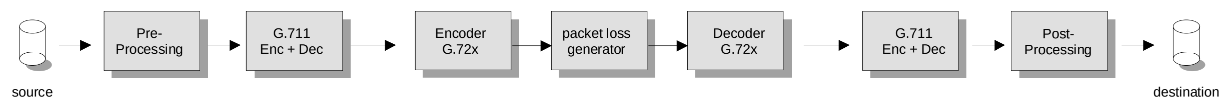
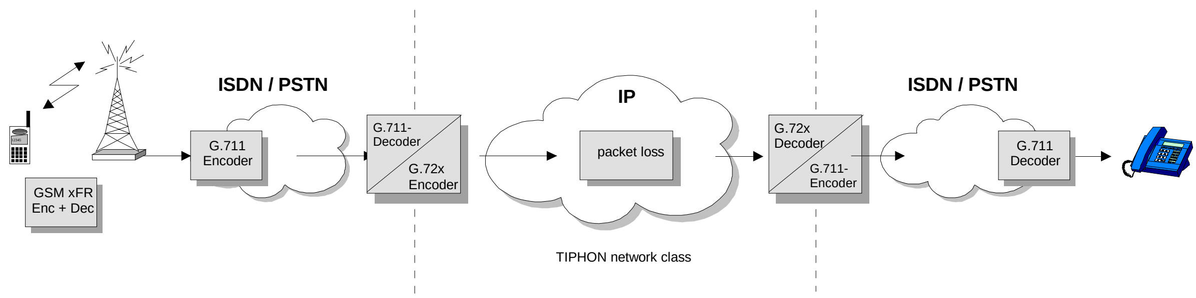


Figure 3: Processing scenario for single codec conditions

Table 3: Tandem Configuration with GSM x FR (Phone to Phone Scenario in including mobile networks)

Tandem with GSM-FR	Packet Loss	Speech Frame Size	Nr. of Frames/ Packet	Substitution
GSM FR + G.723.1 [18] (5.3)	0 %, 5 %, 10 %, 15 %	G.723.1 [18]: 30 ms	2	G.723.1 [18] internal
GSM FR + G.729B [24]	0 %, 5 %, 10 %, 15 %	G.729 [22]: 10 ms	6	G.729 [22] internal
GSM EFR + G.723.1 [18] (5.3)	0 %, 5 %, 10 %	G.723.1 [18]: 30 ms	2	G.723.1 [18] internal
GSM EFR + G.729B [24]	0 %, 5 %, 10 %	G.729 [22]: 10 ms	6	G.729 [22] internal
GSM EFR EP2 + G.723.1 [18] (5.3)	0 %, 5 %, 10 %	G.723.1 [18]: 30 ms	2	G.723.1 [18] internal

logical scenario:



simulation scenario:

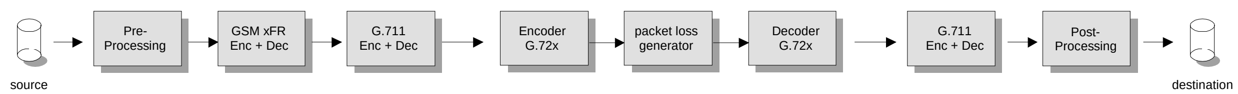


Figure 4: Processing scenario for tandem codec conditions

5.1.1.3 Results

The following figures illustrate the subjective assessment results. On the y axis the MOS score from 1 (bad) to 5 (excellent) is shown. The x-axis shows the various end-to-end scenarios.

5.1.1.3.1 G.711 + Codec + G.711

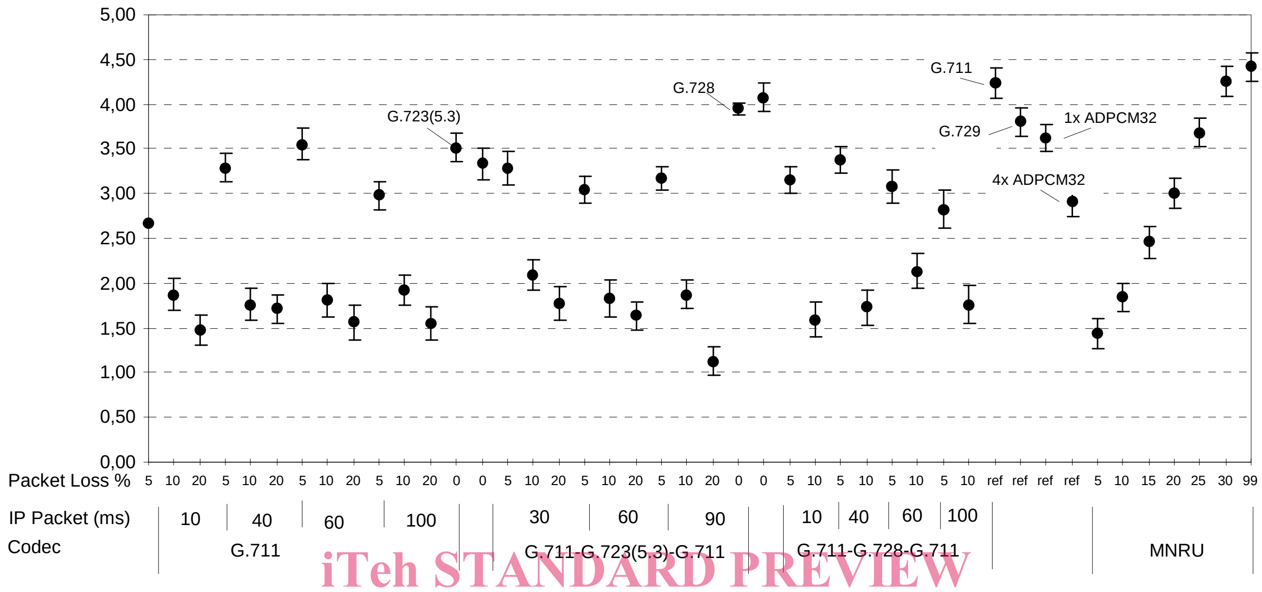


Figure 5: Subjective assessment result

The assessment of single codecs under the influence of packet loss leads to assumptions as follows:

- the packet loss rate of 5 % seems to be almost the quality threshold of MOS 3,0;

- in all test cases the evaluation of voice signals with packet loss of >= 10 % the MOS scores are widely below the quality threshold of MOS 3,0.

5.1.1.3.2 Tandem Conditions with GSM x FR

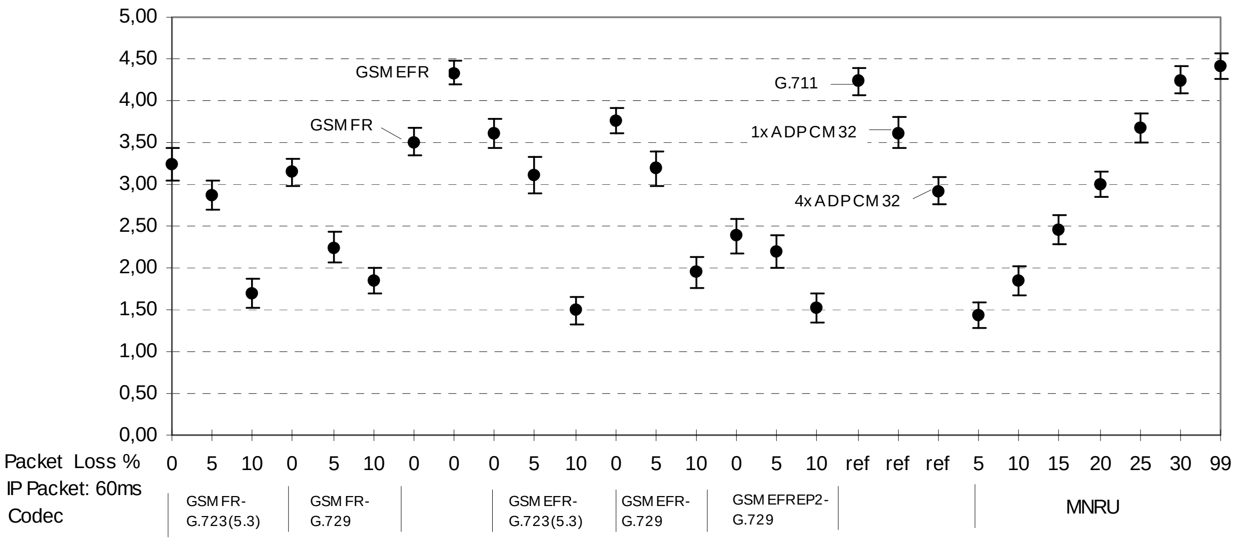


Figure 6: Tandem conditions with GSM x FR